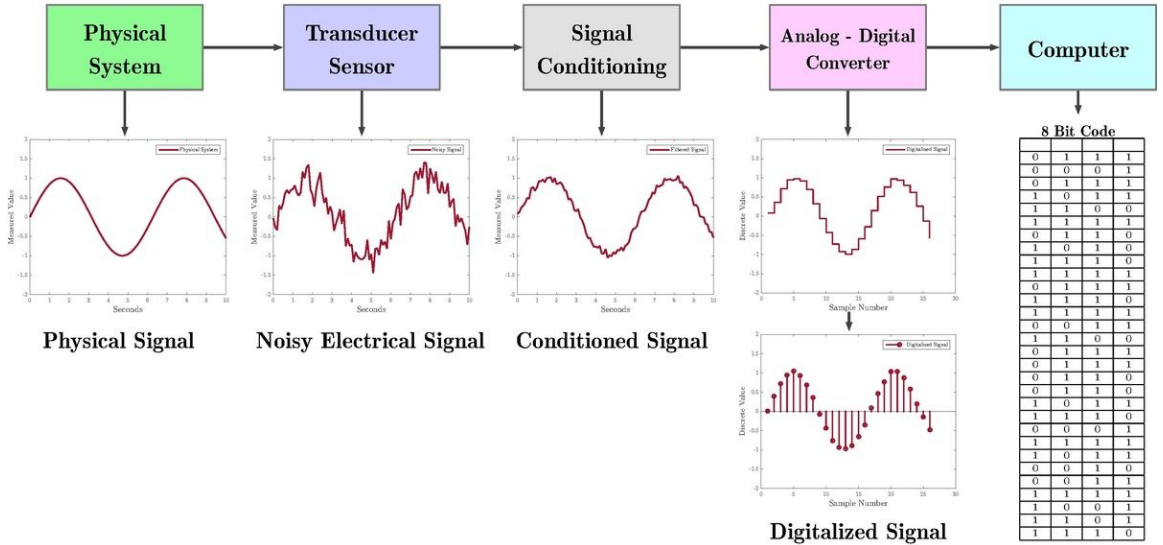
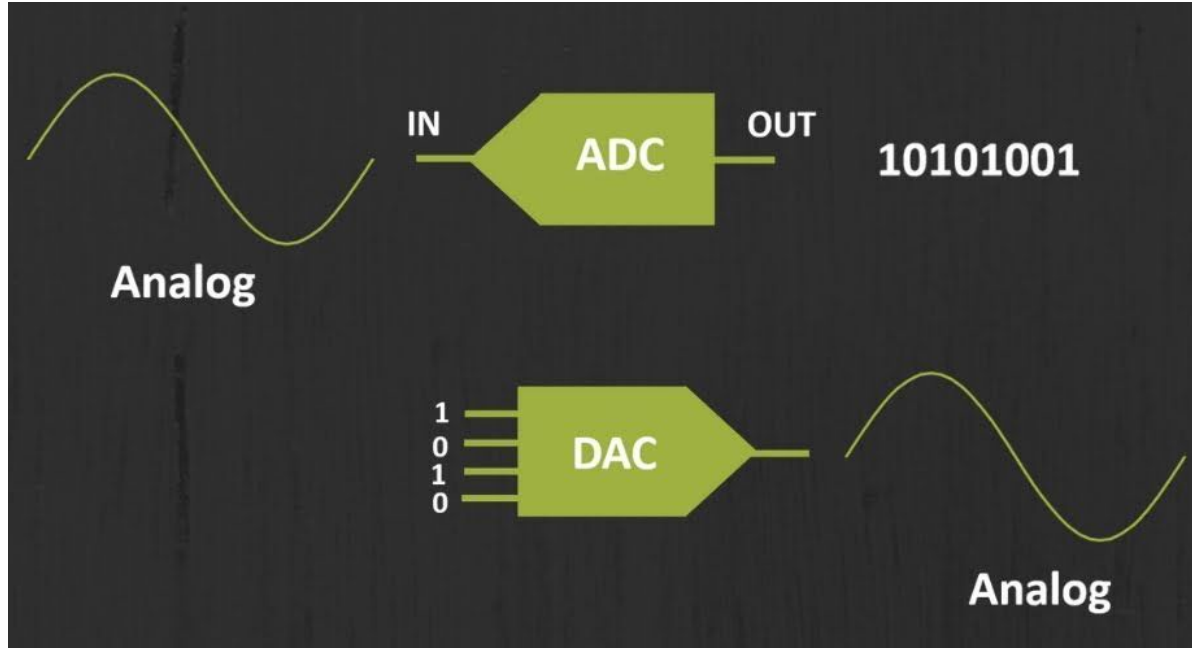


Digital Data Acquisition System

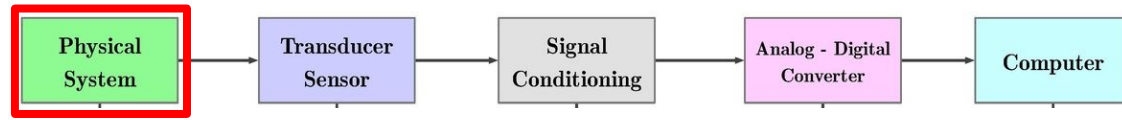


Making a DAQ System that Makes Sense
 John R. Leeman
 GEARS 2023

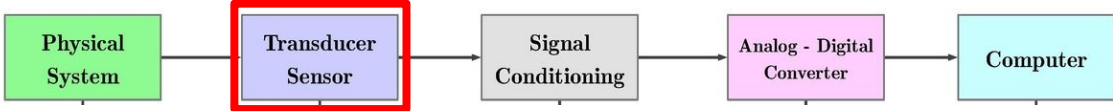
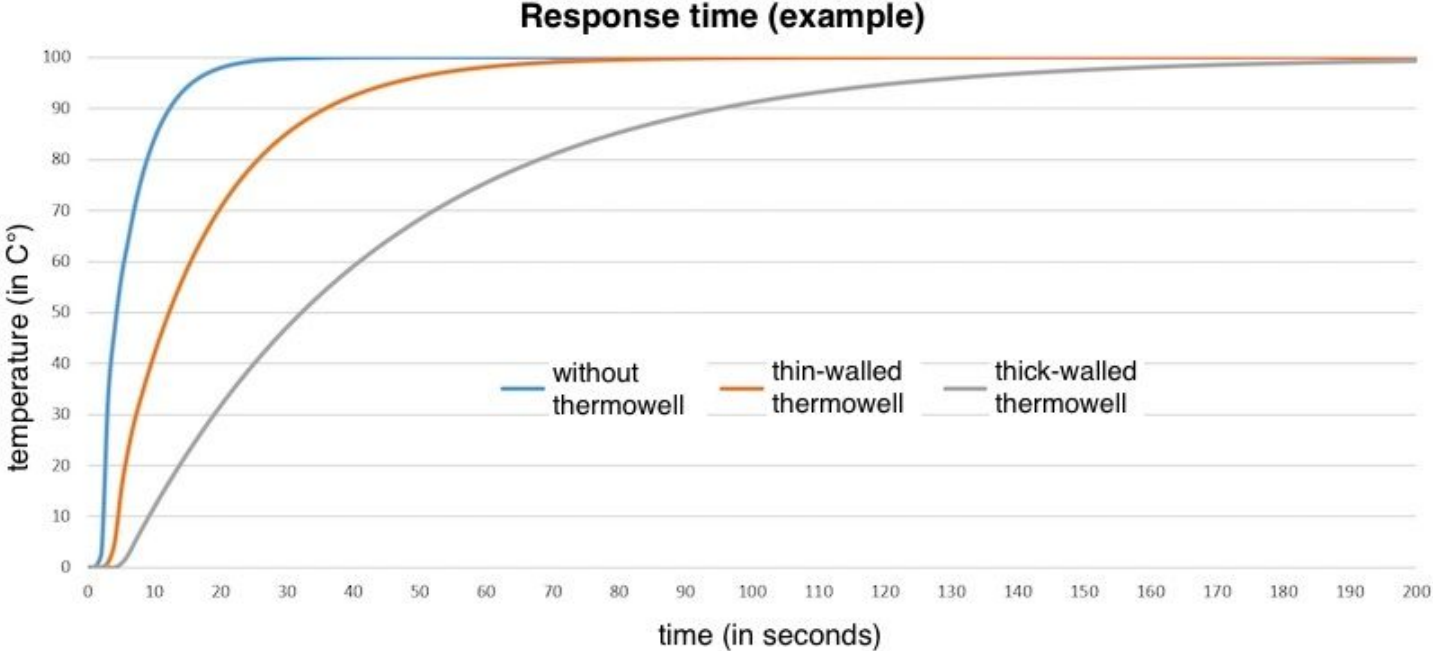
DAQ, ADC, DAC, and more alphabet soup



What you're measuring is likely a filter in itself

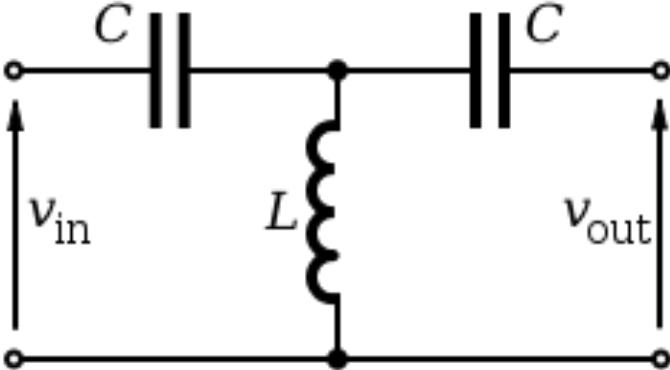


The transducer also certainly has some response function



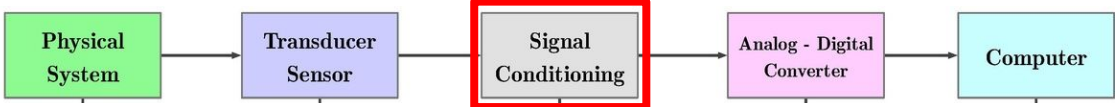
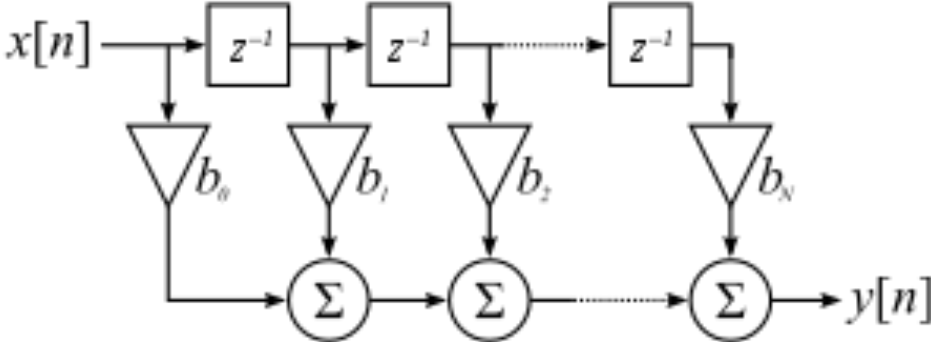
Signal conditioning is amplification and filtering of the signal to get the best SNR possible

Analog Filters



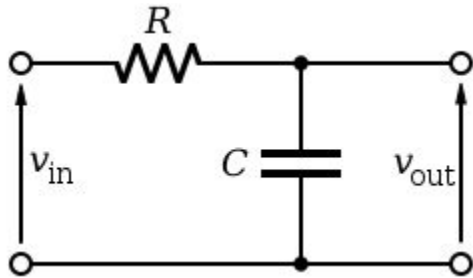
Digital Filters

Image: Wikipedia

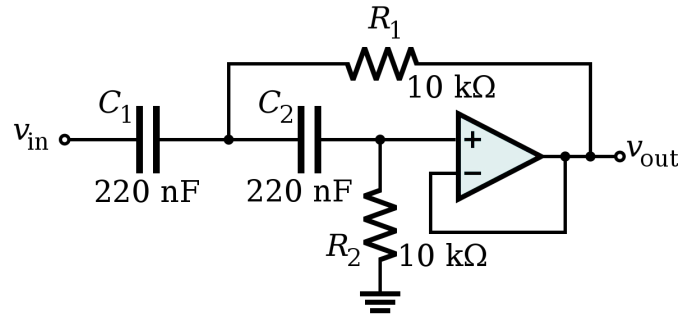


Filters and classed into two main categories with a third “fake” class

Passive



Active



ASIC

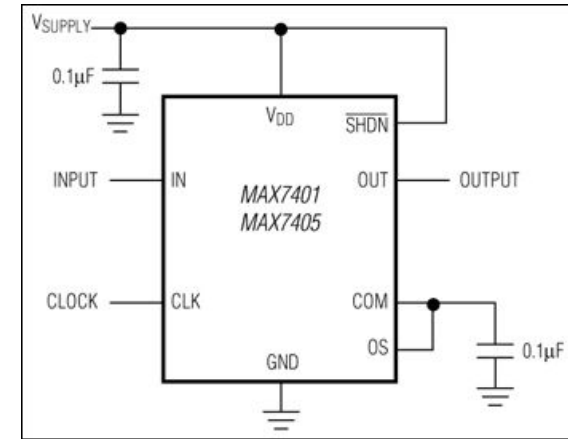
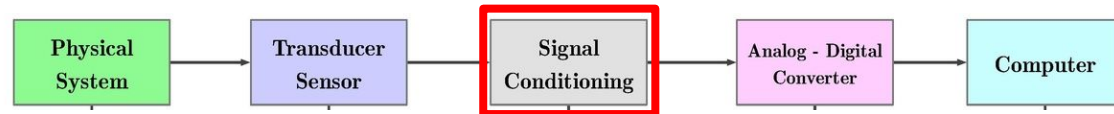


Image: Wikipedia, Maxim Integrated



It is helpful to remember the basic types of filters that we generally employ though

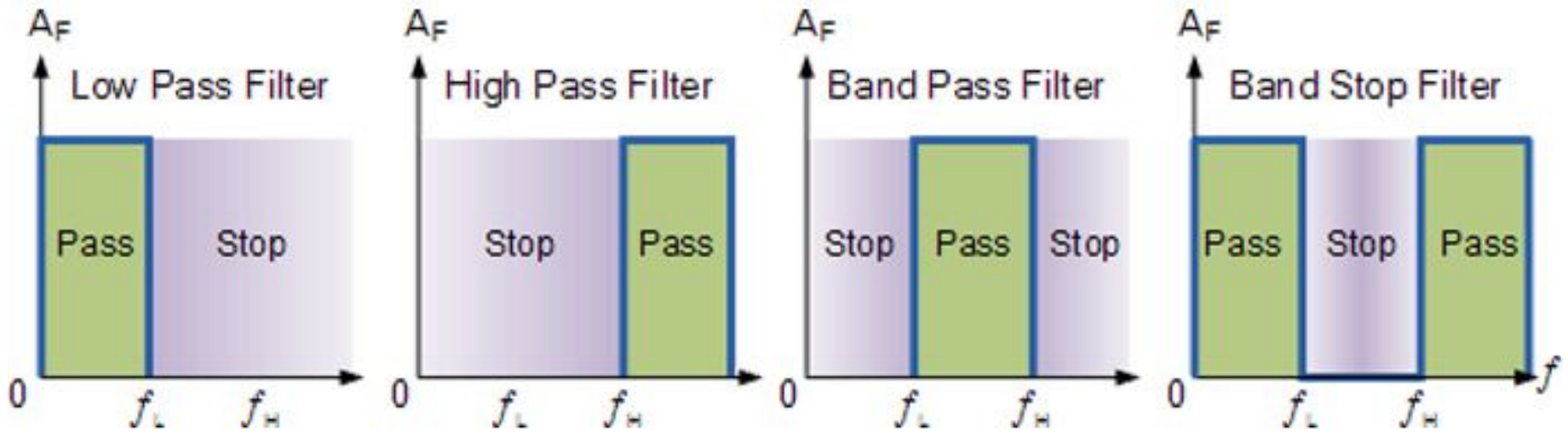
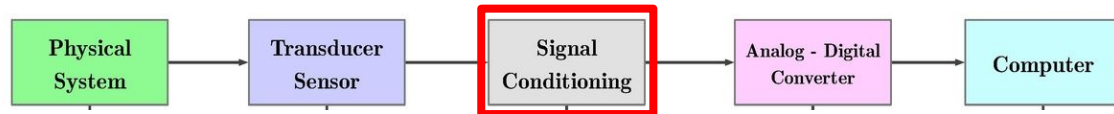


Image: Electronics Tutorials



The most common filter you'll apply is the anti-aliasing filter

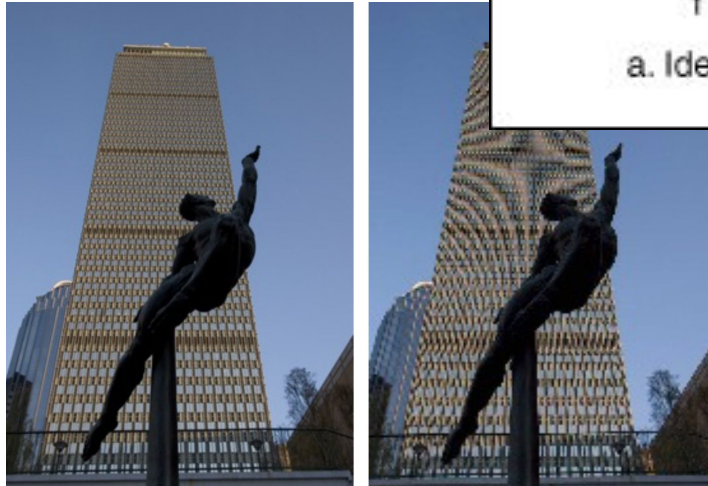
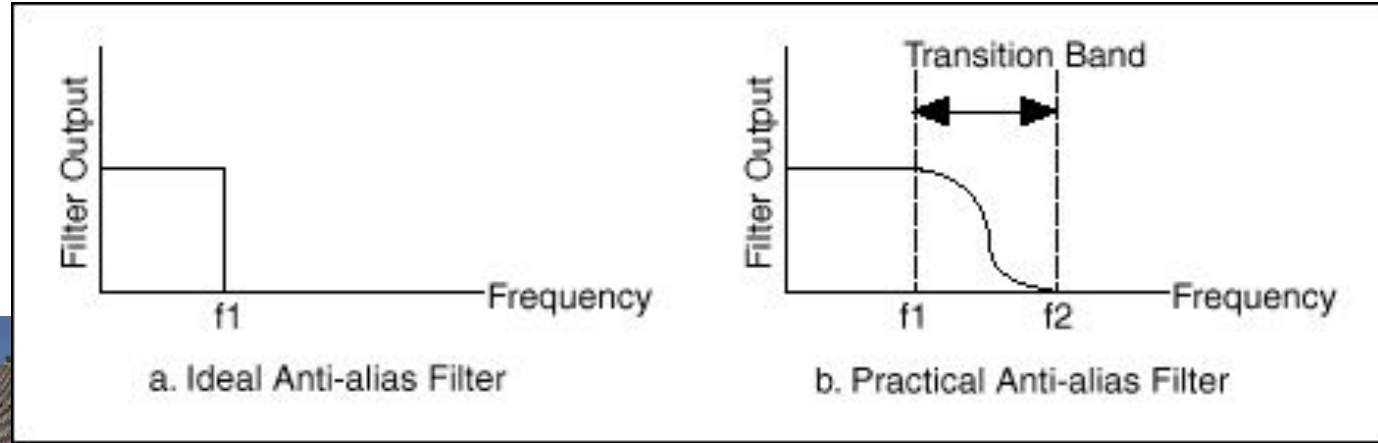
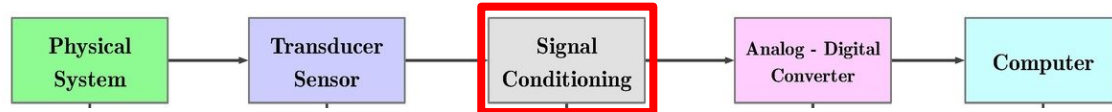
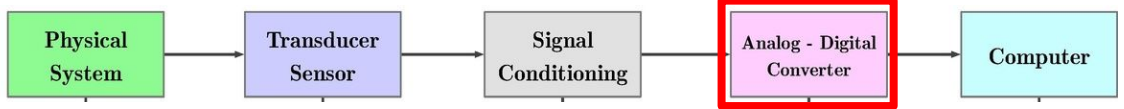
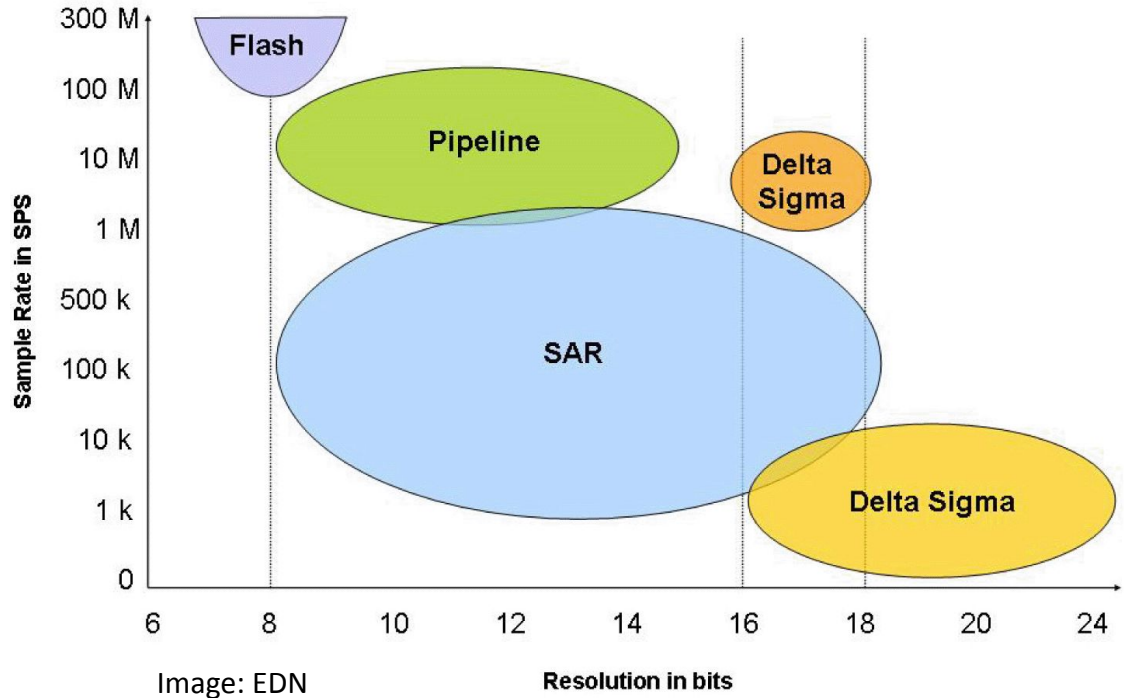


Image: National Instruments, mrpc

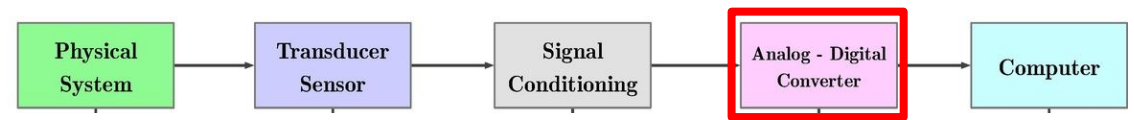


Analog to Digital Converters (ADCs) take a analog (continuous) signal and create a digital representation of it



Let's look at some of the effects of ADC bit depth and sampling rate

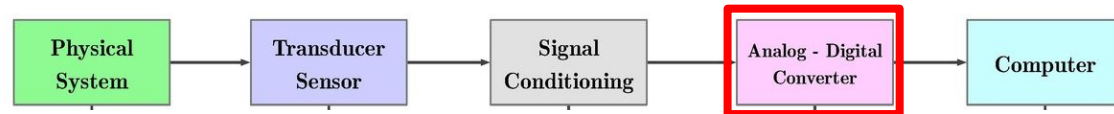
$$\text{volts per bit} = \frac{\text{range}}{2^n - 1}$$



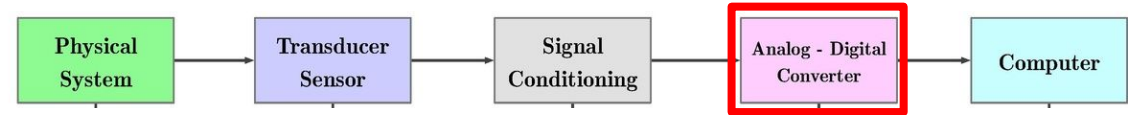
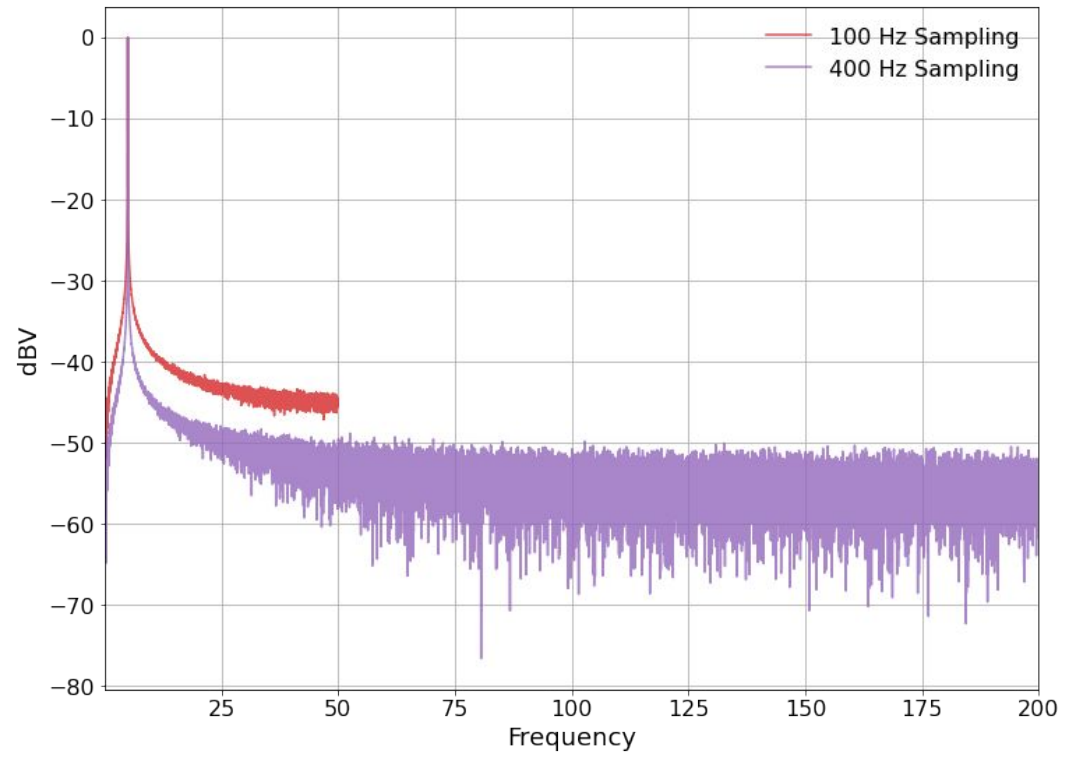
We often look at the SNR and ENOB

$$\text{SNR}_{\text{qe}} = (6.02N_{\text{bits}}) + 1.76\text{dB}$$

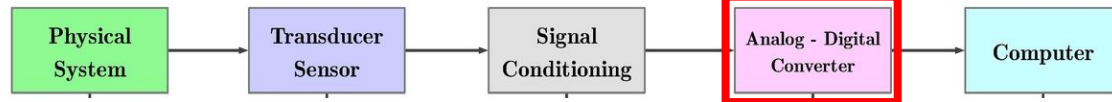
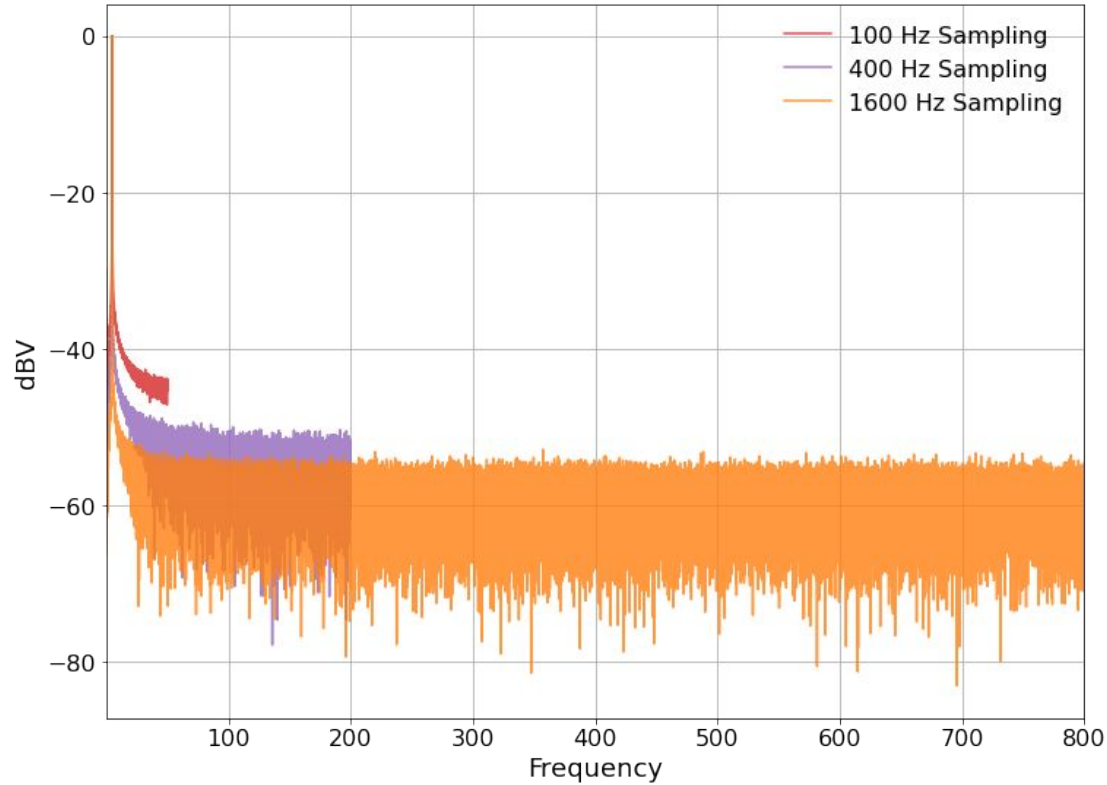
$$\text{ENOB} = \frac{\text{SNR} - 1.76\text{dB}}{6.02\text{dB}}$$



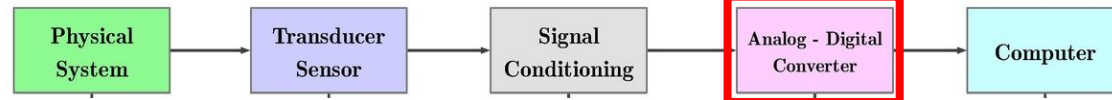
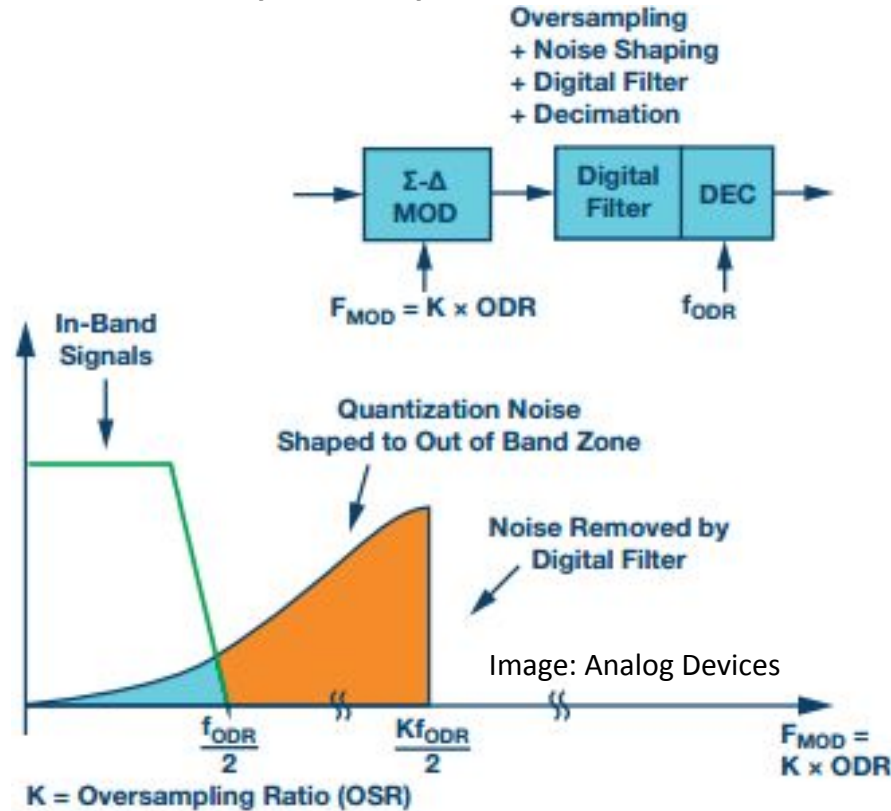
Oversampling is an effective strategy to increase the effective number of bits



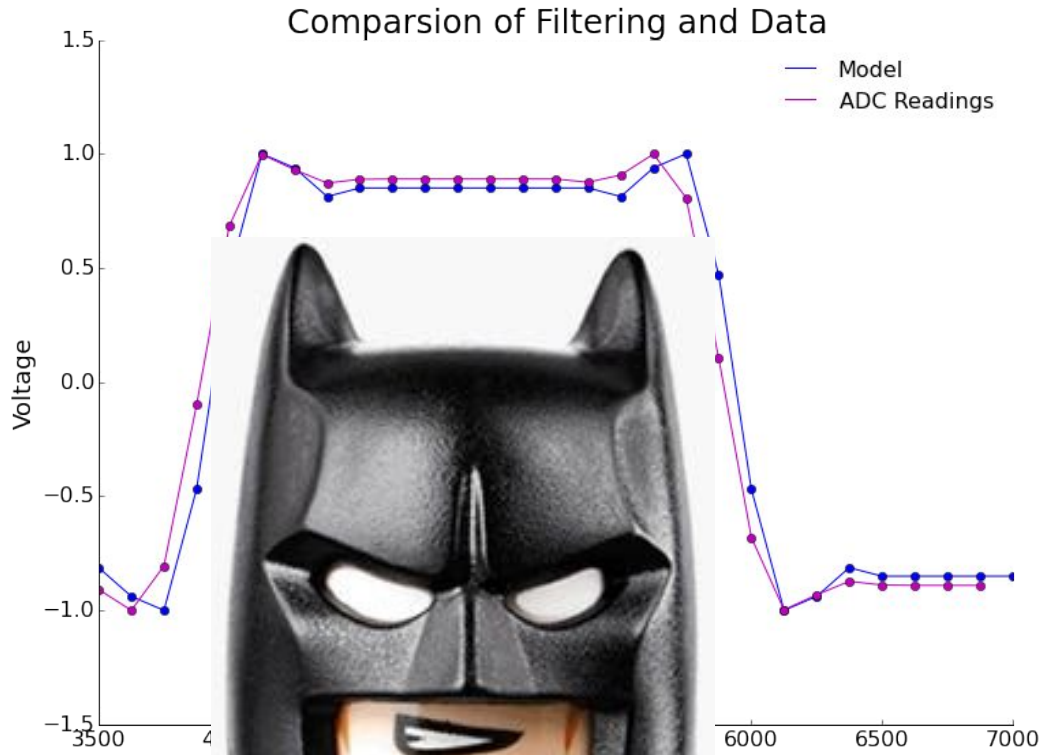
Oversampling is an effective strategy to increase the effective number of bits



Noise shaping is another way many ADCs reduce the noise you see in your signal



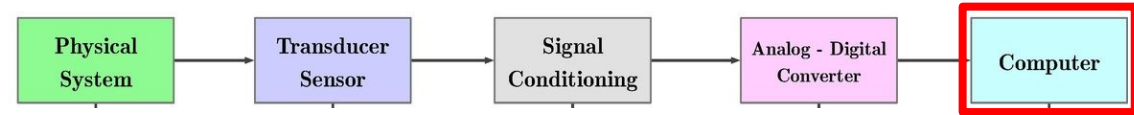
Be careful of filters in the ADC introducing artifacts though!



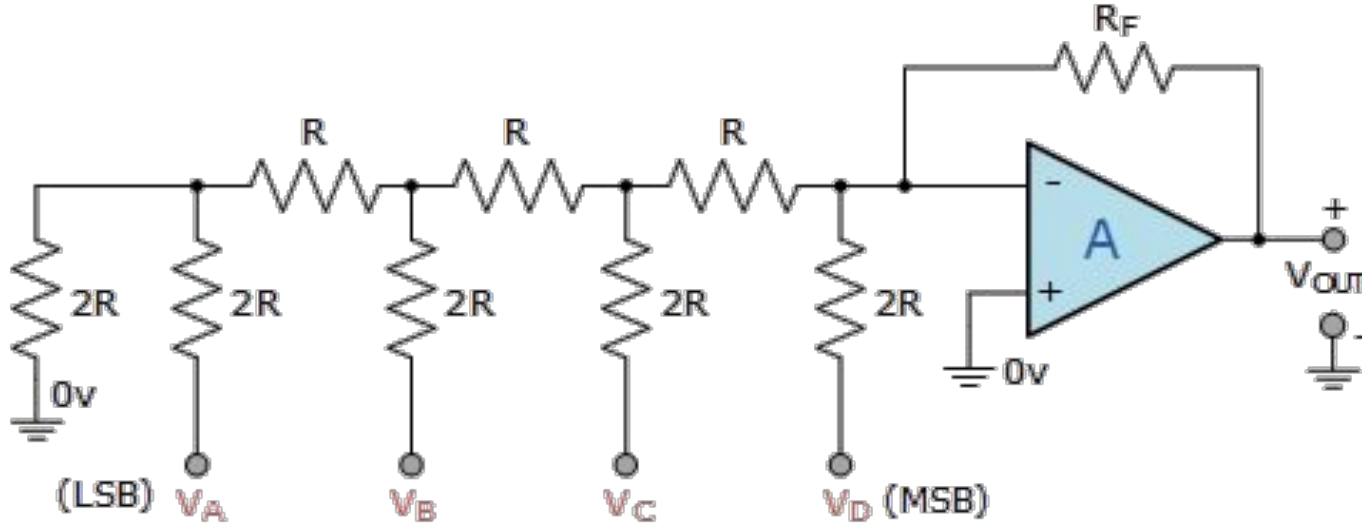
Luckily by the time we digitize we're mostly done with filtering our physical values, but of course we can still drop packets, etc.



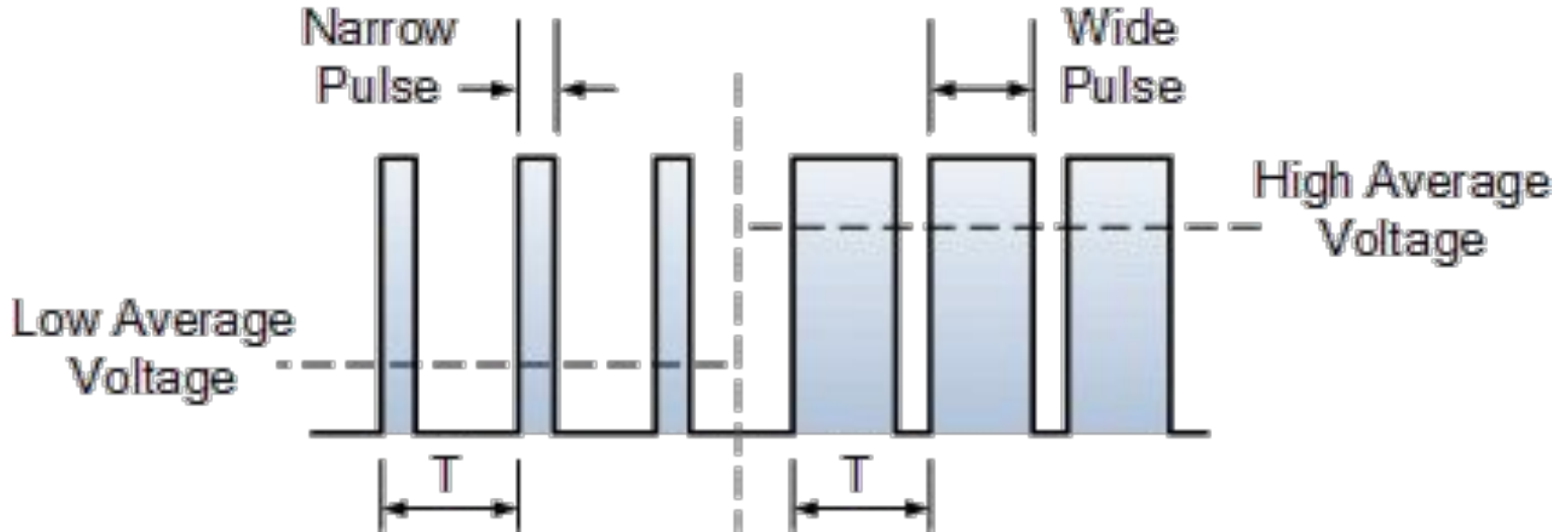
Image: Maxim Integrated



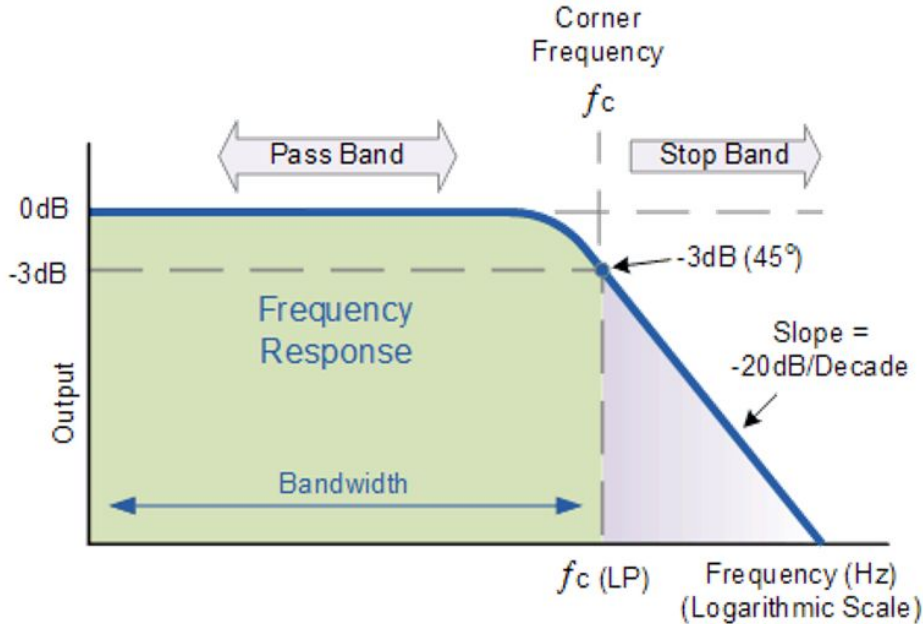
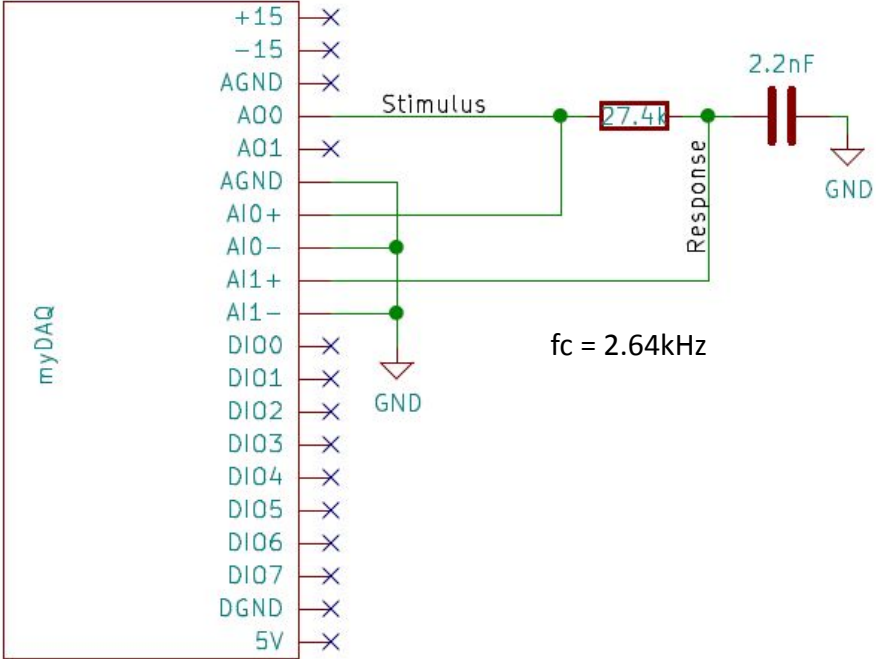
DACs go the other way, creating a voltage for a digital input value



Pulse width modulation

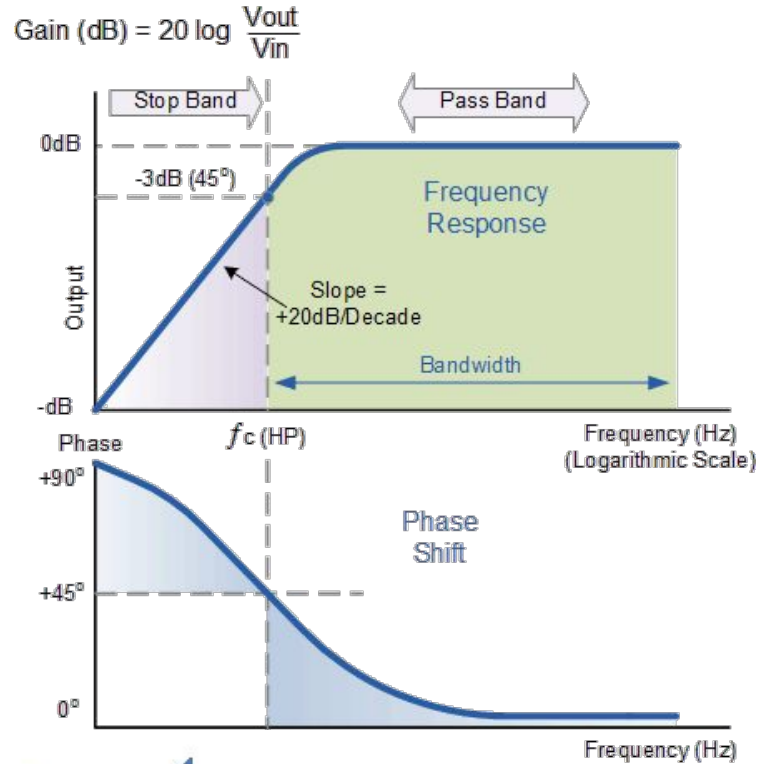
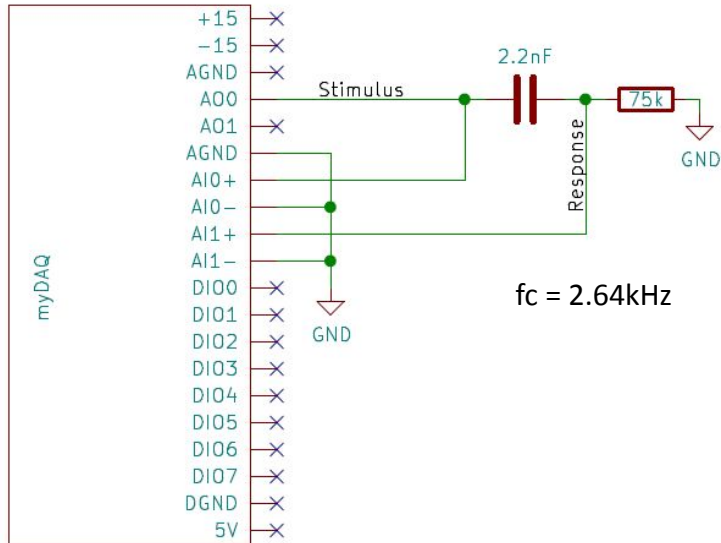


Let's look at the simple RC low-pass filter



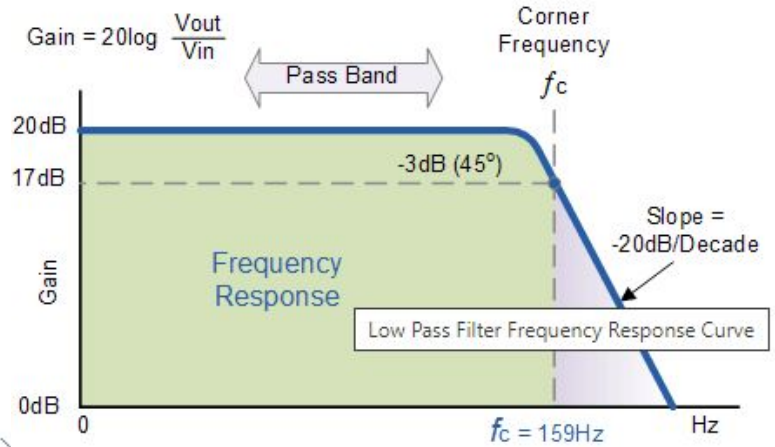
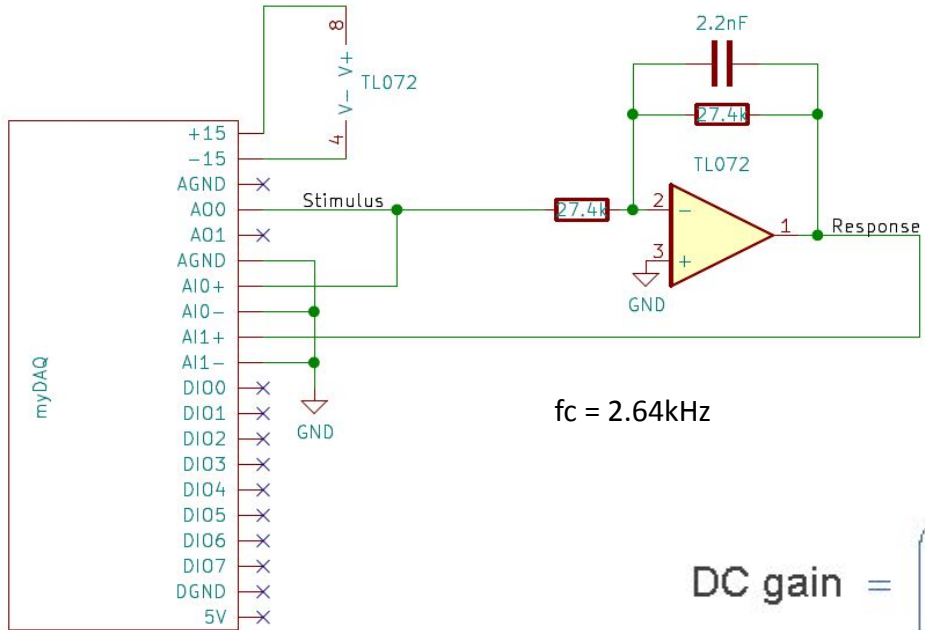
$$f_c = \frac{1}{2\pi RC} \text{ Hz}$$

Now the same thing, but a high-pass



$$f_c = \frac{1}{2\pi RC} \text{ Hz}$$

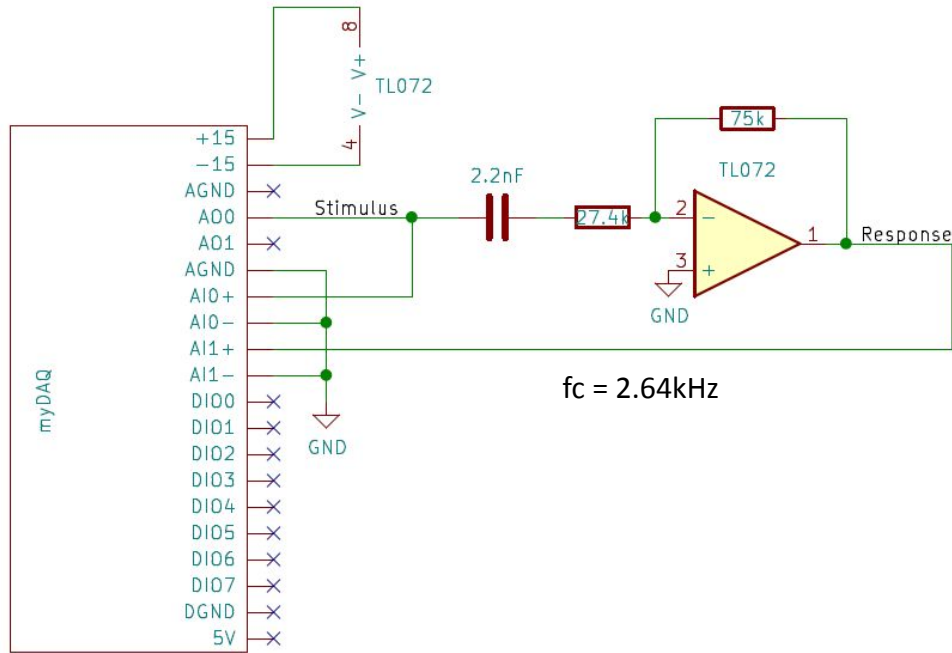
Active filters filter, buffer, and amplify all at the same time



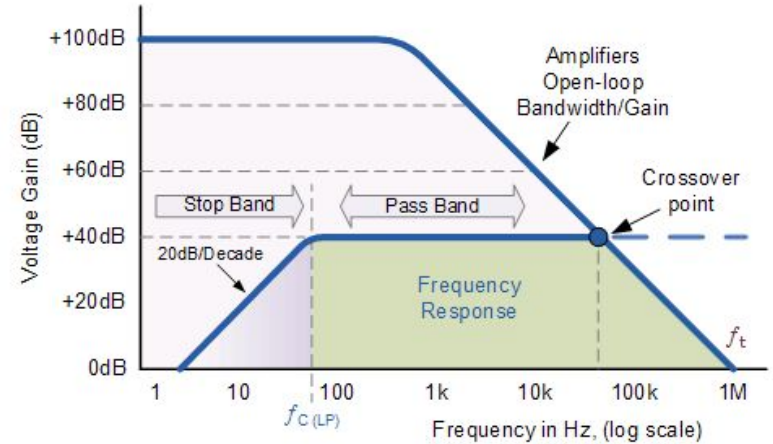
$$\text{DC gain} = \left(1 + \frac{R_2}{R_1} \right)$$

$$f_c = \frac{1}{2\pi RC} \text{ Hz}$$

Active filters filter, buffer, and amplify all at the same time



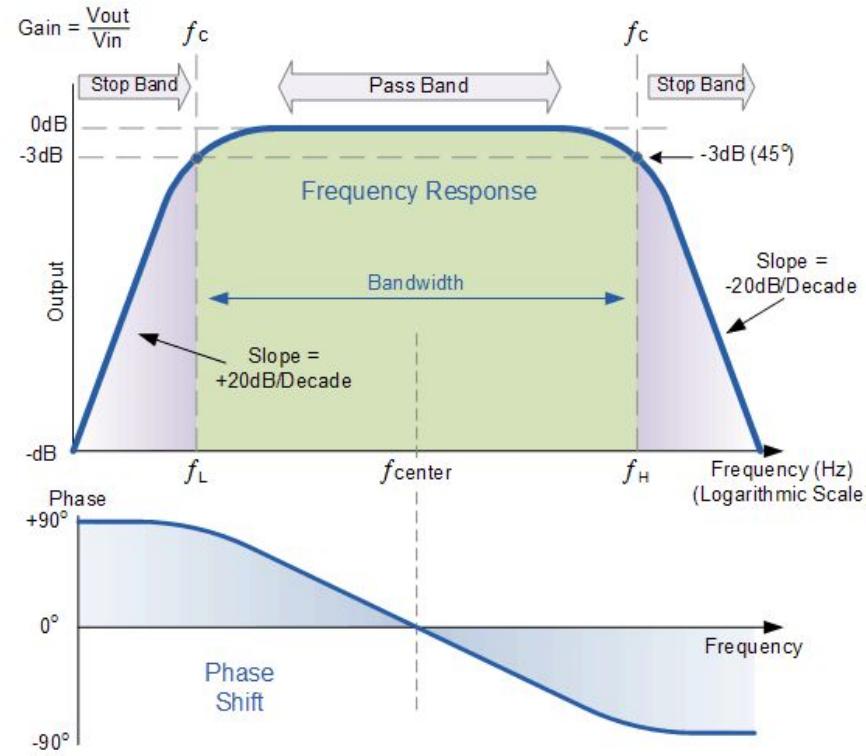
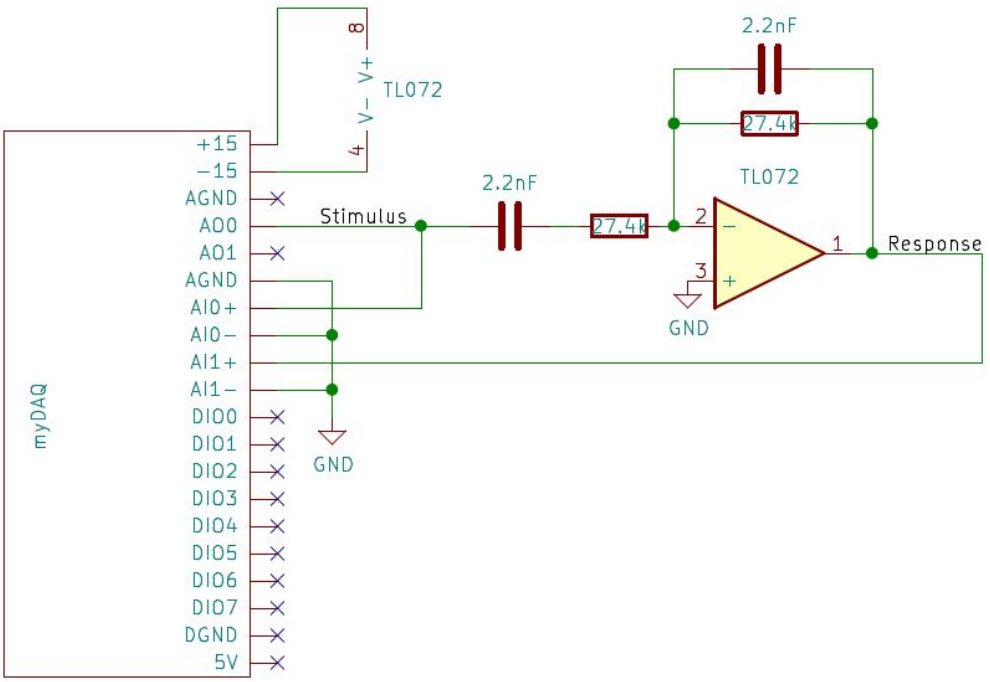
cy Response Curve



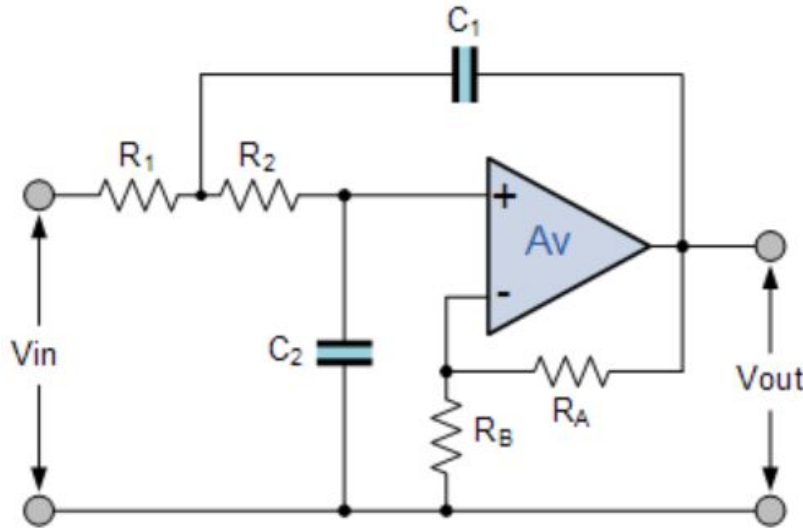
$$A_v = -\frac{R_2}{R_1}$$

$$f_c = \frac{1}{2\pi R_1 C}$$

Combined we can make a bandpass filter



We can continue the complication with higher order filters



$$\text{Gain } (A_v) = 1 + \frac{R_A}{R_B}$$

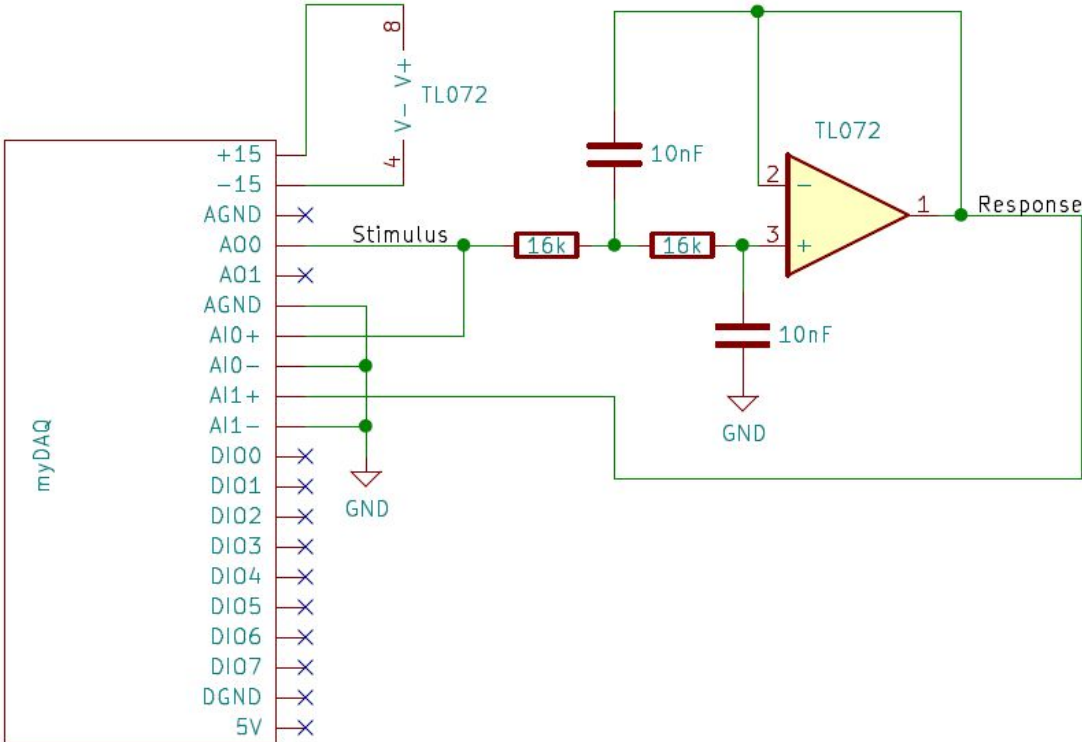
If Resistor and Capacitor values are different:

$$f_c = \frac{1}{2\pi \sqrt{R_1 R_2 C_1 C_2}}$$

If Resistor and Capacitor values are the same:

$$f_c = \frac{1}{2\pi RC}$$

A solid and easy to implement filter is the Sallen-Key



Chip scale filters are really the way to go for sharp and well defined responses, but they are expensive

19-4788; Rev 0; 10/98



8th-Order, Lowpass, Bessel, Switched-Capacitor Filters

General Description

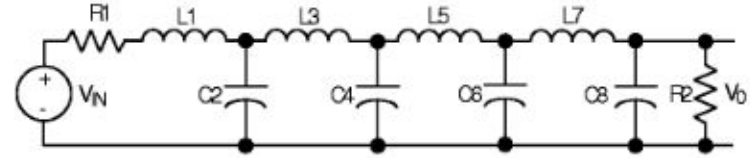
The MAX7401/MAX7405 8th-order, lowpass, Bessel, switched-capacitor filters (SCFs) operate from a single +5V (MAX7401) or +3V (MAX7405) supply. These devices draw only 2mA of supply current and allow corner frequencies from 1Hz to 5kHz, making them ideal for low-power post-DAC filtering and anti-aliasing applications. They feature a shutdown mode, which reduces the supply current to 0.2µA.

Two clocking options are available on these devices: self-clocking (through the use of an external capacitor) or external clocking for tighter corner-frequency control. An offset adjust pin allows for adjustment of the DC output level.

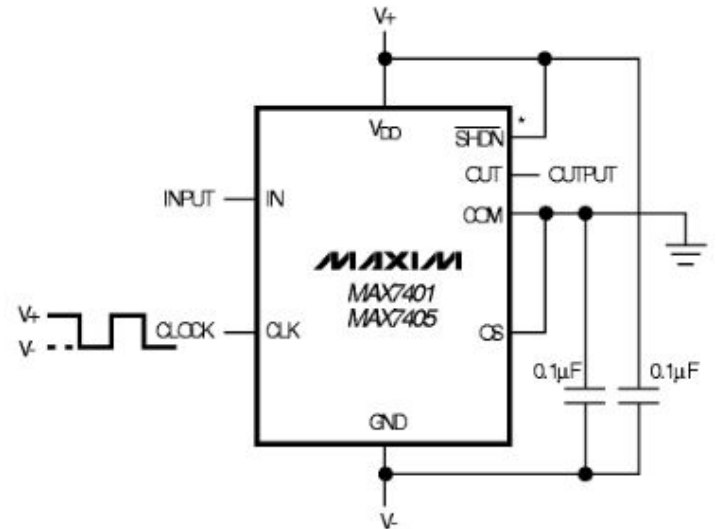
The MAX7401/MAX7405 Bessel filters provide low overshoot and fast settling. Their fixed response simplifies the design task to selecting a clock frequency.

Features

- ◆ 8th-Order, Lowpass Bessel Filters
- ◆ Low Noise and Distortion: -82dB THD + Noise
- ◆ Clock-Tunable Corner Frequency (1Hz to 5kHz)
- ◆ 100:1 Clock-to-Corner Ratio
- ◆ Single-Supply Operation
 - +5V (MAX7401)
 - +3V (MAX7405)
- ◆ Low Power
 - 2mA (Operating Mode)
 - 0.2µA (Shutdown Mode)
- ◆ Available in 8-Pin SO/DIP Packages
- ◆ Low Output Offset: ±5mV



MAX7401/MAX7405



A few other practical notes on filter topologies

- Butterworth - Amplitude accuracy is very flat and well controlled
- Chebyshev - Very steep rolloff, but more ripple in the passband
- Bessel - Uniform time delay for constant group delay (i.e. linear phase response with frequency) and great transient of pulse input response. Have ripple in the passband and slow initial roll-off.

Digital filtering is a different class, but FIR/IIR filters are out there

FIR FILTERS	IIR FILTERS.
$\rightarrow H(z) = \sum_{k=0}^M b_k z^{-k}$	$\rightarrow H(z) = \frac{\sum_{k=0}^M b_k z^{-k}}{\sum_{k=0}^{\infty} a_k z^{-k}}$
\rightarrow FINITE DURATION IMPULSE RESPONSE	\rightarrow INFINITE DURATION IMPULSE RESPONSE.
\rightarrow CAUSAL / CAN BE MADE CAUSAL	\rightarrow NOT GUARANTEED TO BE CAUSAL.
\rightarrow ALWAYS STABLE	\rightarrow NOT GUARANTEED TO BE STABLE.
\rightarrow LINEAR PHASE	\rightarrow NOT LINEAR PHASE.

Checkout <http://t-filter.engineerjs.com/>

There's even an easy Arduino FIR library!

